(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(19) **World Intellectual Property Organization**
International Bureau

(43) **International Publication Date**
19 September 2002 (19.09.2002)

PCT

(10) **International Publication Number**
WO 02/073901 A1

(51) **International Patent Classification⁷:** **H04L 12/413**, 12/56

(21) **International Application Number:** PCT/SE01/00512

(22) **International Filing Date:** 9 March 2001 (09.03.2001)

(25) **Filing Language:** English

(26) **Publication Language:** English

(71) **Applicant** *(for all designated States except US)*: **TELEFONAKTIEBOLAGET LM ERICSSON (publ)** [SE/SE]; S-126 25 Stockholm (SE).

(72) **Inventor; and**
(75) **Inventor/Applicant** *(for US only)*: **MALOMSOKY, Szabolcs** [HU/HU]; Szècheyi ter 31, H-2000 Szentendse (HU).

(74) **Agent: MAGNUSSON, Monica**; Ericsson Radio Systems AB, Patent Unit Radio Access, S-164 80 Stockholm (SE).

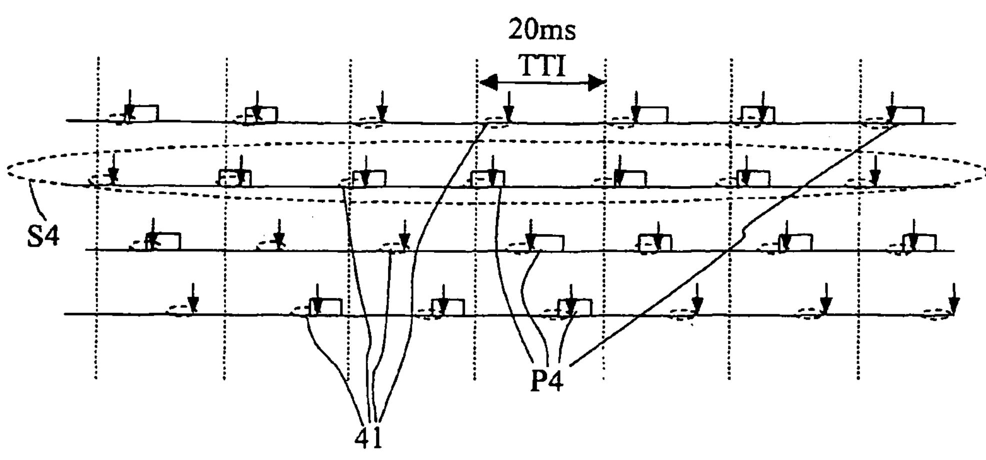
(81) **Designated States** *(national)*: AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DZ, EE, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MA, MD, MG, MK, MN, MW, MX, MZ, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, SL, TJ, TM, TR, TT, TZ, UA, UG, US, UZ, VN, YU, ZA, ZW.

(84) **Designated States** *(regional)*: ARIPO patent (GH, GM, KE, LS, MW, MZ, SD, SL, SZ, TZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE, TR), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GW, ML, MR, NE, SN, TD, TG).

Published:
— *with international search report*

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

(54) **Title:** METHOD AND DEVICE FOR TRANSMISSION OF PACKET DATA TRAFFIC ON A MULTIPLEXED LINK



(57) **Abstract:** The present invention relate to packet data transmission between nodes in a radio access system. The packets are scheduled periodically and smoothed over time by applying a corresponding phase offset (42) for each connection. A problem addressed is how to reduce packet losses on the links. It is observed that the arrivals of the packet streams at receiving node have a bursty character. The present invention reduces the burstiness by introducing a small random delay to each packet. The delay causes the packets to jitter on the link.

WO 02/073901 A1

METHOD AND DEVICE FOR TRANSMISSION OF PACKET DATA TRAFFIC ON A MULTIPLEXED LINK

TECHNICAL FIELD OF THE INVENTION

The present invention relates to transmission of information packets in a communication system for example a radio access network and especially to a method for transmission of packet streams and to a node handling such streams.

DESCRIPTION OF RELATED ART

UTRAN is a short name of the UMTS (Universal Mobile Telecommunication System) Radio Access Network. A mobile terminal MT, a radio base station BS and a radio network controller RNC are three basic nodes in UTRAN, and shown in figure 1. The mobile terminal MT is connected by a radio link to the radio base station BS. From the radio base station BS radio links with several mobile terminals MT are provided. The radio network controller RNC is connected to several radio base stations BS by fixed links, however in figure 1 just one exemplar of each of the nodes is disclosed. Communication in the direction to the mobile terminal MT, i.e. in the downlink direction, is set up via the radio network controller RNC and via the radio base station BS to the mobile terminal MT. The communication is also conveyed from the communication source via an exchange connected to the radio network controller RNC, however, no nodes except the radio network controller RNC, the radio base station BS and the mobile terminal MT will be described here. Communications in the opposite direction, i.e. in the uplink direction, is transmitted via the same nodes.

Figure 2 shows protocol stacks up to layer 2/3 of the radio network controller RNC, the radio base station BS and the

mobile terminal MT. The protocol stack shows layers of protocols that correspond to specific functionalities for establishing communication.

Also shown in figure 1 are the fixed link interface Iub between the radio network controller RNC and the radio base station BS, and the air interface Uu between the radio base station BS and the mobile terminal MT.

The lower layers of the protocol stacks are related to physical and functional requirements of transmitting data from the sending to the receiving node. The lowest layer on both sides of the air interface Uu is thus WCDMA (Wideband Code Division Multiple Access) that corresponds to the radio multiple access technology. In the radio network controller RNC in the Iub side of the radio base station BS the transport network infrastructure L1 is depicted, e.g. an ATM/AAL2 network or an IP network. Any of these is used for transferring MAC frames.

The MAC frames are produced by the above MAC (Media Access Control) layer. The MAC layer as well as the RLC (Radio Link Control) layer are located in the radio network controller RNC and in the mobile terminal MT.

The radio links shall be utilised optimally because the capacity of these radio links is limited.

Streams of information packets in the downlink direction are formatted into streams of radio frames with a certain length and format in the radio network controller RNC. The main function of the MAC layer in the downlink is to schedule the radio frames. The scheduling of the radio frames is done periodically in every 10, 20, 40 or 80 ms to adapt to the radio interface.

There are strict QoS (Quality of Service) requirements on the Iub interface. This applies also to interfaces between

different radio network controllers RNC, however this interface is not shown in figure 1. The QoS requirements are strict both on delays and packet losses.

Consider a general packet communication receiving process in order to discuss the QoS requirements. Various connections 33 (shown in figure 3) are multiplexed in a queuing system, which consists of a FIFO (First In First Out) buffer 32 of size b and a server 31 with rate S. The queuing delay packets suffer when waiting in the buffer must be below a certain limit. As long as the packets of the received flows are evenly distributed over time and the total flow rate are not higher than the serving rate of the server 31, all packets will be served the moment they arrive. No queuing delays will thus arise. If, however, the traffic flow is bursty, i.e. the packets are unevenly distributed over time, while the packet flow rate is the same on a time averaged basis, packets will have to be queued in the buffer 32 before being served. The more uneven distribution of the packets, the longer the buffer 32 need to be. Packets in the buffer 32 will be delayed, the longer queue of packets the longer delay. Packets delayed longer than maximum are deleted and thereby lost.

One way of reducing the delays would of course be to increase the capacity of the server 31. Another known way to reduce the delay in the buffer, and thereby losses, is to shape the traffic according to e.g. the well known Leaky-bucket method. However, a Leaky-bucket like shaper is not applicable in UTRAN because multimedia traffic, and especially voice traffic can not be a subject of such shaping because its real-time character requires low delays. Leaky-bucket like shaping is usually applied to non-real time traffic to modify the packet arrival patterns such that it conforms to a simple, mathematically tractable traffic description.

Figure 4a shows the multiplexing of packet streams on the transport network L1 of fixed interface Iub. In this example there are four voice connections, each corresponding to a packet stream on one of the horizontal lines.

The multiplexing also corresponds to the order in which the packets will arrive at the fixed interface Iub. The packets are scheduled periodically with a constant interarrival time TTI (Transmission Time Interval). In figure 4a TTIs are indicated by vertical dotted lines. The TTI are 20 ms long for voice, however, different services have different TTIs. The example with voice service on the Iub link can be generalised to all types of services and also to other links.

When a new voice connection is established, a phase offset 42 is selected for the connection. The phase offset 42 is selected randomly, according to a uniform distribution, over the TTI. The objective of the phase offset 42 selection is to smooth the traffic and thereby decrease burstiness.

The packet transmission is started after the phase offset 42 is selected. The packet transmission can be modelled with a periodic on-off source. Packets are transmitted during the "on-periods" and not transmitted during the "off-periods". In figures 3, 4a and 5, small boxes indicate packet transmissions in the on-periods.

Figure 5 shows four connections 53 with periodic packet arrivals multiplexed in a queuing system, which consists of a server 51 and a FIFO buffer 53. The buffer temporarily stores the arrived packets if the server 53 is busy when they arrive. The buffer depth b corresponds to the maximum delay on the fixed interface Iub as required by the QoS. If the buffer is full when a packet arrives the packet will be lost.

SUMMARY OF THE INVENTION

The present invention addresses a problem that packets are lost when periodic scheduling is used on a transmission link.

The solution to this problem is based on the observation that even though a phase offset is randomly selected for each packet stream, in order to smooth the traffic, the transmissions of multiplexed packets on the link may nevertheless have a bursty character. When the packets at the end of a burst are received the receiving server is busy and the packets are delayed in the server buffer and possibly lost. Moreover, since the selected phase offset is used for the whole transmission of a stream of packets, all packets of the stream may be received in an unfavourable time with respect to the server queue.

The present invention solves the problem by a method introducing a random delay to each packet in the transmitting node. The delay is randomly selected within an interval and causes the burstiness to decrease and and the arrival order of packets of different streams to vary. Thereby the maximum server queue is reduced and, moreover, the delay will be more fairly distributed over the various packet streams.

The problem is also solved by a node for an electronic information system comprising an input for receiving information streams and a packet output link and that is arranged to transmit the information streams as packet streams scheduled periodically on the packet link. The transmitter is further arranged to introduce a small randomly selected delay to each packet, and thereby causing the packets to jitter on the link.

The invention is also solved by a shaper arranged to introduce a s random delay to periodically scheduled packets

that are to be transmitted on a multiplexed link as streams of packet data.

The invention has the great advantage of reducing the packet losses without having to increase the receiving server capacity.

A further advantage is that losses are distributed more fairly over the streams.

A further advantage is that the invention is easy to implement with respect to the protocol stack used in communication systems such as UTRAN. The functions to introduce the random delays will then be implemented between those handled by the MAC-layer and those handled by the Transport Network Layer.

DESCRIPTION OF THE DRAWINGS

Figure 1 is a block diagram of basic nodes in a radio access system.

Figure 2 is a block diagram of UTRAN protocol layers.

Figure 3 is a block diagram showing packet streams fed to an input buffer of a receiving server.

Figure 4a is a diagram of scheduling of 4 packet streams on a link according to prior art.

Figure 4b is the same as figure 4a but with the invention applied.

Figure 5 is a diagram corresponding to that of figure 3 but with the packets scheduled periodically and time divided into transmission intervals.

Figure 6 is a flow chart showing the introduction of a random delay on the packets.

Figure 7 is a time diagram showing phase relations of the packets of 4 different streams.

Figure 8 is a a block diagram of a packet data traffic shaper.

Figure 9 is a diagram showing packet losses as a function of the maximum length of the random delay introduced.

Figure 10 is a diagram showing the gain with respect to non introduced delay as a function of maximum length of the random delay introduced.

DESCRIPTION OF PREFERRED EMBODIMENTS

The present invention is based on the observation that the prior art scheduling imposes a bursty character of the packet arrivals. Figure 7 shows the prior art phase relations of 4 packets of different streams that are scheduled and multiplexed on the same link. Each packet stream corresponds to a connection, i.e. a voice connection or a multimedia connection. A packet P1-P4 from each of the stream is transmitted within the same TTI (Transmission Time Interval). One of the packets P3 overlap in time with two other packets P3, P4. Assume that the server 51 has capacity for handling just one packet P1-P4 at the time. The third and the fourth packets P3, P4 will be buffered before being served and the fourth packet P4 will be buffered the longest time. The phase relations remain the same for subsequent TTIs, all with respect to prior art.

According to the present invention a small random delay is introduced to each packet before the packets are multiplexed on the fixed interface Iub. The phase relations between the packets will thus differ between subsequent TTIs and the burstiness will decrease. The independence of the different packet streams is maintained.

Figure 4b is the same as figure 4a except for that the transmission moment for the packets P4 are slightly varying between different TTIs as a result of the random delay introduced to the packets P. The random delay interval 41 is also indicated in figure 4b.

Figure 6 is a flowchart showing the steps performed by different protocol layers of a transmitting node, in the UMTS system. In a first step 61, the packets of various streams are scheduled periodically and deterministic according to the principle disclosed with reference to figure 4a. In short this means that there is an opportunity for each stream to have a packet scheduled at a precise occasion within each TTI. The precise moment is determined by the phase offset 42 of each packet stream. The scheduling is handled by the MAC-layer of the protocol stack shown in figure 2.

In a next step 62 a small random delay is introduced to each packet P4. The delay is randomly selected from an interval 41 that is much smaller than the TTI.

In a following last step 63, the packets P4 are multiplexed on the link. This is handled by the Transport Network layer, i.e. layer 1 of the protocol stack shown in figure 2.

The small delay introduced in step 62 causes packets to jitter on the link. By jitter is meant that the arrival times differs slightly and randomly from the expected arrival times. The jitter will be in the region of [phase offset; phase offset + X] where X denotes the maximum delay.

To implement the invention just the second step 62 is added.

To perform the random-delay shaping step 62, a random-delay shaper is implemented. Figure 8 discloses the shaper 81 that comprises a random number generator 82, a table 83, a clock 85, an input from the MAC layer, an output to the transport

layer and a central processor 86 that controls the functioning of the Shaper 81.

Consider downlink traffic. After MAC frames are encapsulated into Iub frames and sent towards the transport layer, they are first entering the random-delay shaper 81. The shaper 81 attaches a temporary identifier to the MAC frames and timestamps the arriving MAC frames. The random number generator 82 generates random numbers according to a uniform distribution from the interval 41 [0ms ; Xms], X being the maximum introduced delay. A random delay is generated for each MAC frame and the random delay value is added to the timestamp of the MAC frame. The temporary frame identifiers and the related timestamps (updated with the random delay) are then inserted into the table 83, which is ordered according to the timestamp values. The MAC frames are stored in the buffer 84. If the internal clock 85 of the shaper (81) reaches the value of the first timestamp in the table (83), the MAC frame with the corresponding temporary identifier is sent to the transport network.

As a consequence of QoS requirements on maximum total delay on the fixed interface Iub, the receiving server buffer 51 depth need to be reduced correspondingly to the maximum random selected delay introduced. This could suggest that the advantages with the introduced random delay would not compensate for the disadvantage of the corresponding reduction of the receiving buffer depth 51. Consider the following example in order to evaluate the transmission performance with the invention applied:

In a voice connection, compressed voice packets are scheduled from the MAC layer every 20 ms (TTI), they can be delayed at most 5 ms on the Iub interface, and the average packet loss should be at most around 0.1% - 1%.

387 voice connections, i.e. packet streams, are multiplexed over a 2720 kbit/sec link. The voice connections have random phase offsets 42. Voice packets are sent periodically in the 20ms TTIs according to an on-off process. The length of the on period is exponentially distributed with a mean of 10 sec. The probability of the on period is 0.5. The codec is a 12.2 kbps codec, it sends 31 bytes in the on periods. In the off periods, 8 bytes are sent in every 8^{th} TTI.

The parameters of the voice model are not very important in our evaluation. The most important is that the voice and other services in UTRAN can be modelled as an on-off source with deterministic arrivals.

The voice packets are multiplexed in an AAL2 multiplexer and transported in ATM cells to the Base Station.

In a diagram of figure 9, the loss performance results are depicted, for various maximum lengths of the random delay applied to each packet. On the Iub interfaces there is a maximum 5ms-delay requirement. The horizontal axis of the diagram corresponds to the buffer 51 size that was decreased from 5ms to 1ms, while the uniform distribution range controlling the randomisation was increased from [0ms; 0ms] to [0ms; 4ms]. So, for example for a 3ms buffer 51, the random delay introduced in the Random Delay Shaper step 72 was chosen according to a uniform distribution over the range [0ms; 2ms]. This way, the maximum 5ms-delay requirement was met in all cases. The uniform black line shows the average loss and the dashed line shows the maximum loss for the various buffer depths.

At all 9 points in the curve, 10 simulations were run in a simulator. Each time ~800000 packets were simulated.

For the maximum buffer 51 depth of 5 ms, no random delay at all was applied and the curves for 5 ms buffer depth thus represents the performance of state of the art. From figure

9 can be seen that introducing only a small delay e.g. [0ms; 1ms] at 4ms buffer results already in a large decrease both in the average loss, above 1% to 0.5%, and in the maximum loss, 28% to around 6%. The maximum and the average are calculated over the 10 simulation runs and over the 387 connections.

What happened is that the average delay including the randomisation delay increased, while the burstiness of the arrival's process has been reduced. This way, the losses got lower. At the same time, since the randomisation results in a constantly varying phase relation arrival order on different connections, the packet losses are distributed more fairly over the connections.

Increasing the average delay is acceptable, since the interesting performance measure is not the average but the tail probability e.g. 99% quantile of the delay.

To make it easier to quantify the advantages of the solution in the case of this example the achieved gains are showed in Figure 10. The uniform black line shows the gain related to the average packet loss (AVGGAIN) while the dashed line shows the gain regarding maximum packet loss (MAXGAIN), compared to applying a buffer 51 depth of 5 ms and no random delay. The horizontal axis corresponds both to the buffer size and to the maximum random delay, in the respect that the maximum total delay of both shall be 5 ms. For a buffer size of 5 ms there is thus no random delay at all applied to the packets. From the curve it can be concluded that a buffer size of 3 ms and [0ms; 2ms] random delay, the maximum loss became 7 times smaller than with no random delay and 5 ms buffer 51 depth.

CLAIMS

1. A method of transmitting (73) a number of packet data streams (S4) on a multiplexed link (53) wherein the packets (P4) of said packet data streams are scheduled periodically (71), c h a r a c t e r i s e d in,

introducing (72) a randomly selected delay (42) to each packet (P4) and thereby causing the transmission time of the packets in said streams to jitter.

2. The method of claim 1 wherein an offset (41) is allocated to each of said packet data streams, said offset defining a moment of possible transmission of the packets (P4) of respective data stream (S4).

3. The method of claim 3 wherein said offset (42) defines the moment for transmission within transmission time intervals (TTI).

4. The method of claim 3 wherein said randomly selected delay is smaller than the transmission time interval (TTI).

5. The method of claim 4 wherein the said delay is selected according to a uniform distribution in a delay interval (41) [0;X] (41), X being the maximum introduced delay.

6. A method according to claim 1 wherein periodically scheduled packets (P4), and a timestamp of their respective scheduling time plus the selected random delay are buffered with reference to the corresponding packet and wherein the packets are read from the buffer in order to be multiplexed on the link when a clock time corresponds to the value of the respective timestamp plus the selected random delay.

7. The method of claim 1 or 3 wherein the scheduling is handled by a MAC protocol layer and the multiplexing by a Transport Network layer.

8. A node (RNC, BS) for an electronic information network comprising,

an input for receiving a number of information streams (S4),

an output (Iub) to a packet link (53), and

arranged to transmit the received information streams as corresponding packet data streams (S4) scheduled periodically and multiplexed on the output link,

charcterized by,

shaping means (81) for introducing a random delay (41) to each packet (P4) and thereby cause the transmitted packets to jitter.

9. A node according to claim 8 arranged to allot a phase offset (42) to each packet data stream, said phase offset defining a possible transmission moment within transmission time intervals (TTI).

10. A node for en electronic information system arranged to carry out the method of claim 1,2,3,4,5,6 or 7.

11. A node according to claim 8 or 10 that is a radio base station (BS) or a radio network controller (RNC).

12. A radio access network arranged to carry out a method as claimed in claim 1, 2, 3, 4, 5, 6 or 7.

13. A radio access network comprising a node as claimed in claim 8, 9 or 10.

14. A packet data traffic shaper (81) comprising,

an input for receiving streams (S4) of periodically scheduled packets (P4),

an output for transmission on a multiplexed link,

a random delay selector (82) for selecting a random delay for each packet,

a buffer (83,84) for buffering the received packets and an arrival time plus the selected delay for the respective packet,

control means (86) comprising a clock (85) and arranged to control the buffer, the delay selector and the clock and arranged to read packets from the buffer to the output when the respective arrival time plus selected delay corresponds to the clock time.

15. A node for en electronic information system including a shaper as claimed in claim 12.

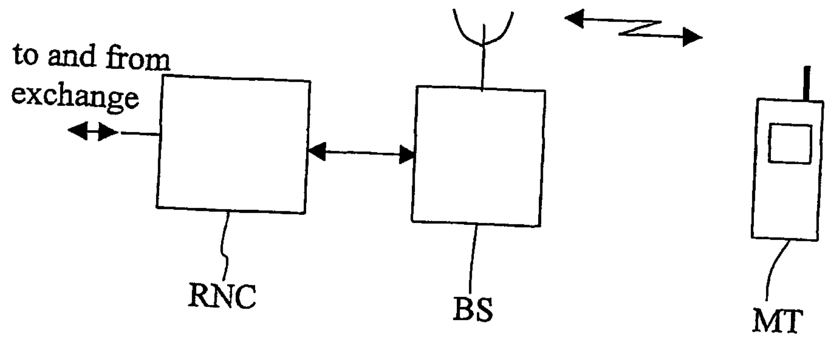

to and from
exchange
RNC
BS
MT

Fig. 1

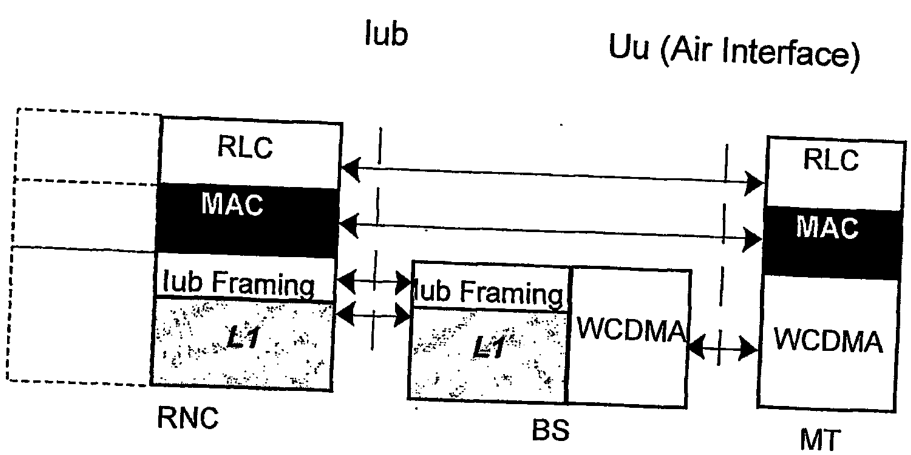

Iub
Uu (Air Interface)
RLC
MAC
Iub Framing
L1
Iub Framing
L1
WCDMA
RLC
MAC
WCDMA
RNC
BS
MT

Fig. 2

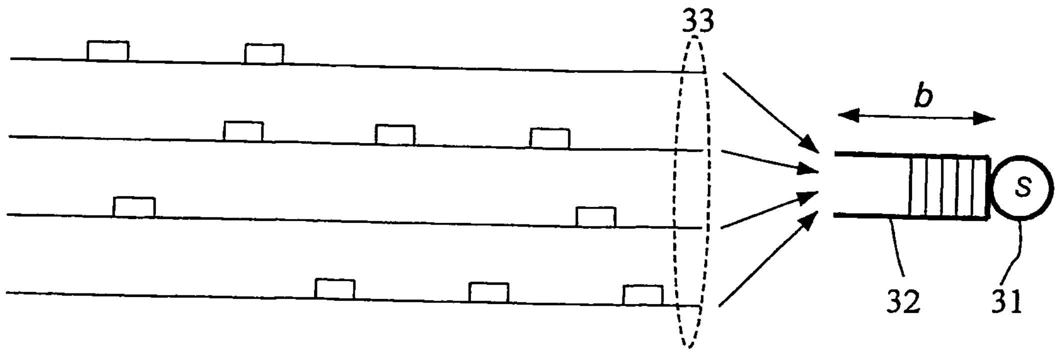


Fig. 3

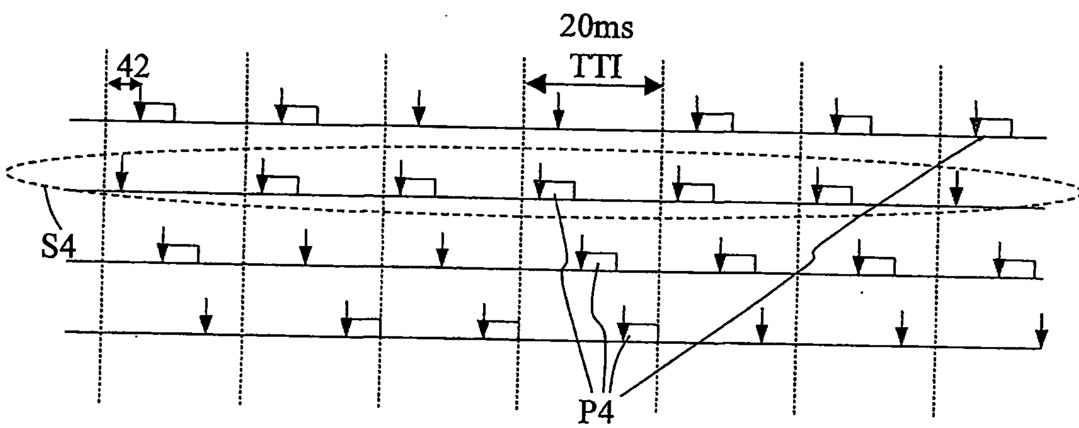


Fig. 4a

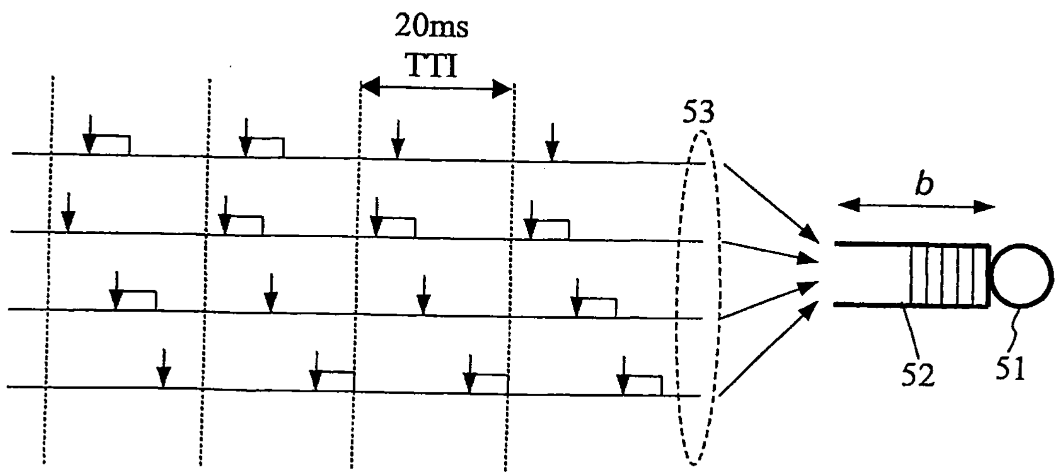


Fig. 5

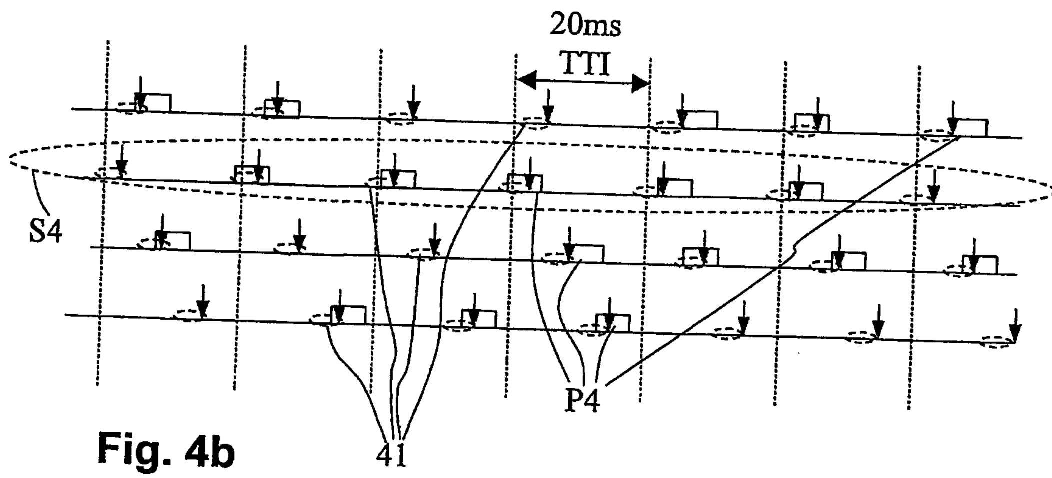


Fig. 4b

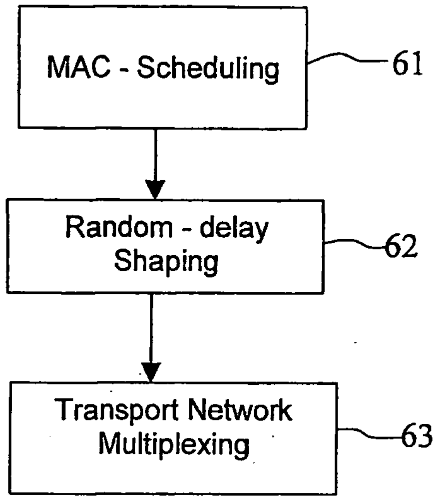

MAC - Scheduling
61
Random - delay
Shaping
62
Transport Network
Multiplexing
63

Fig. 6

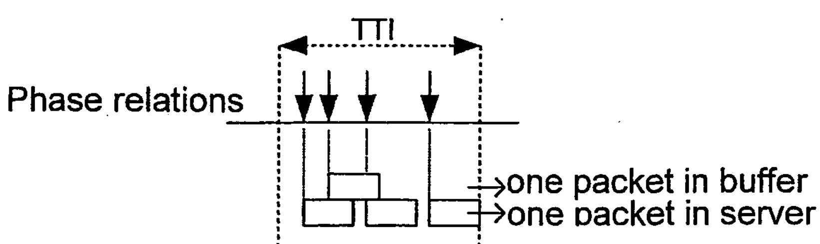

TTI
Phase relations
one packet in buffer
one packet in server

Fig. 7

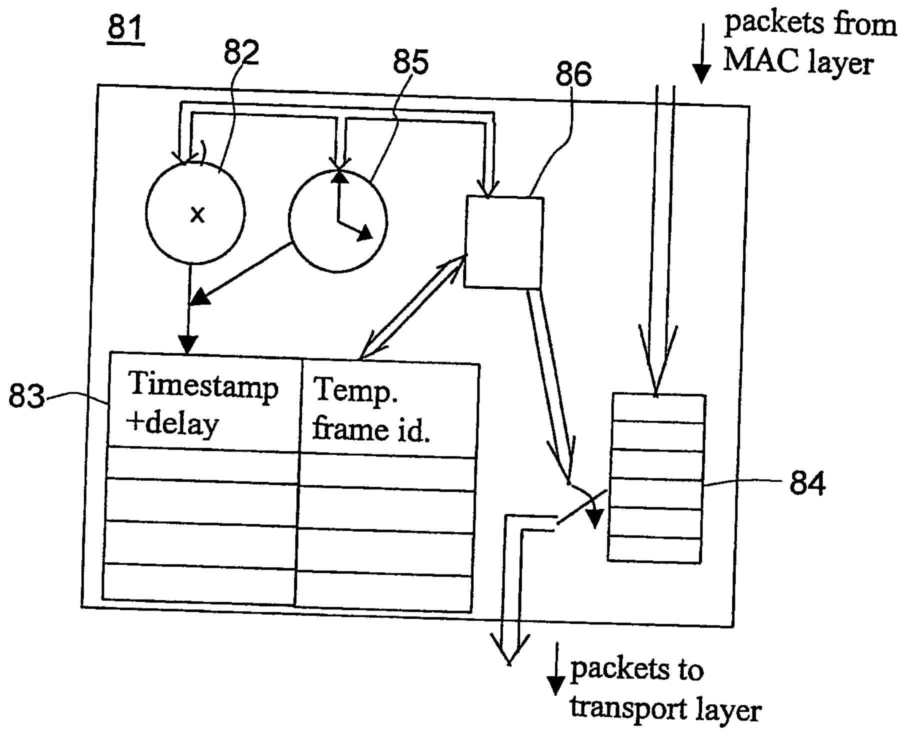

81
82
85
86
packets from MAC layer
x
Timestamp +delay
Temp. frame id.
83
84
packets to transport layer

Fig.8

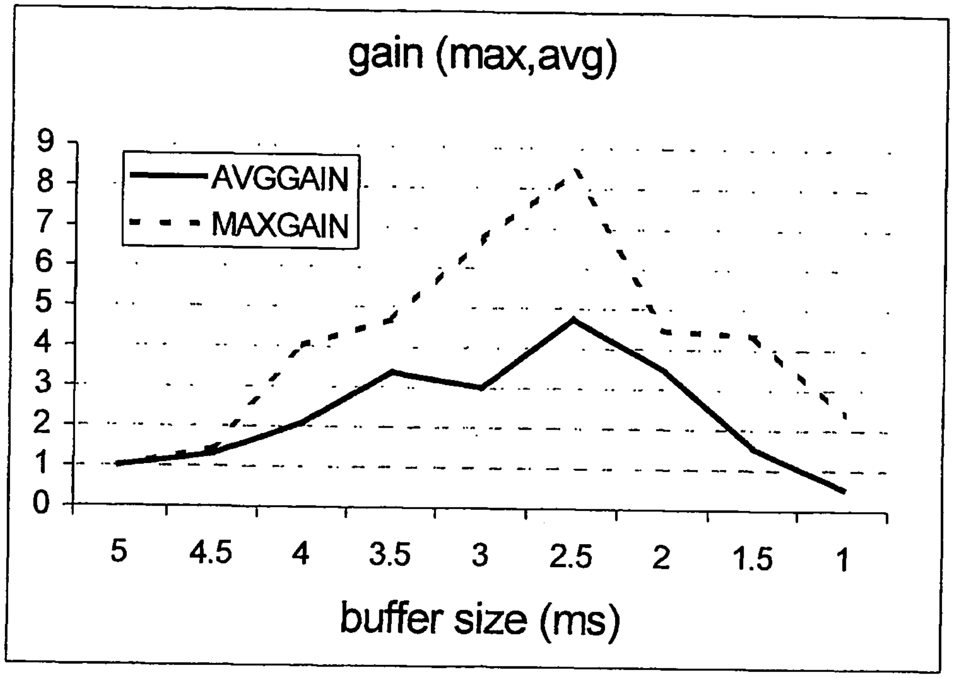

gain (max,avg)
AVGGAIN
MAXGAIN
9
8
7
6
5
4
3
2
1
0
5
4.5
4
3.5
3
2.5
2
1.5
1
buffer size (ms)

Fig. 9

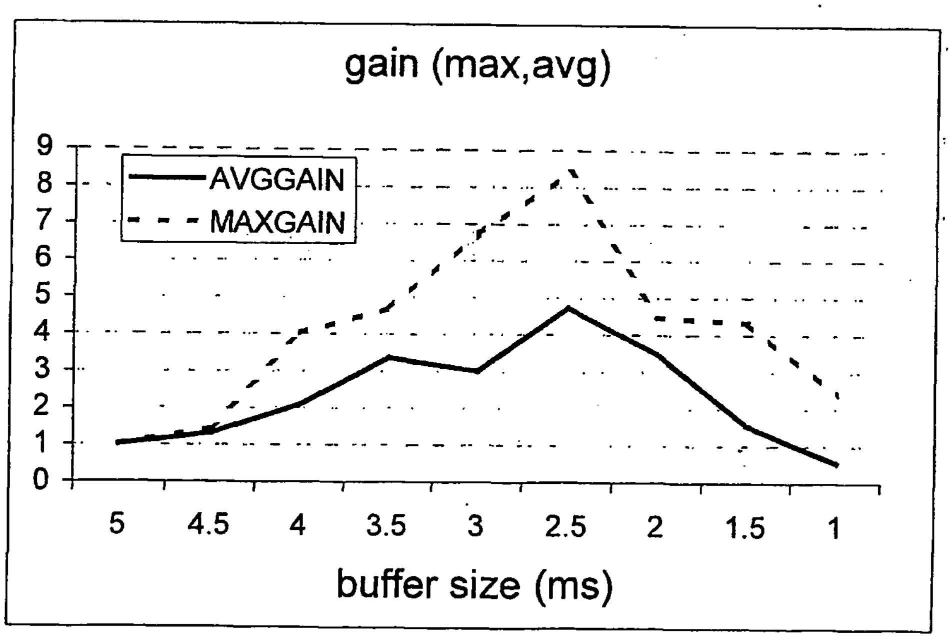

gain (max,avg)
AVGGAIN
MAXGAIN
9
8
7
6
5
4
3
2
1
0
5
4.5
4
3.5
3
2.5
2
1.5
1
buffer size (ms)

Fig. 10

INTERNATIONAL SEARCH REPORT

International application No.
PCT/SE 01/00512

A. CLASSIFICATION OF SUBJECT MATTER

IPC7: H04L 12/413, H04L 12/56

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC7: H04L, H04Q

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

SE,DK,FI,NO classes as above

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

EPODOC,WPI,IEL

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 6055578 A (R.A.WILLIAMS ET AL), 25 April 2000 (25.04.00), column 7, line 4 - line 25; column 14, line 12 - line 26 --	1-15
A	US 5854900 A (M.KALKUNTE ET AL.), 29 December 1998 (29.12.98), column 2, line 44 - line 64 --	1-15
A	US 5850525 A (M.KALKUNTE ET AL), 15 December 1998 (15.12.98), column 3, line 46 - line 50; column 7, line 28 - line 34 --	1-15
A	US 4412326 A (J.O.LIMB), 25 October 1983 (25.10.83), abstract --	1-15

[X] Further documents are listed in the continuation of Box C. [X] See patent family annex.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier application or patent but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance: the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance: the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
10 October 2001	12-10-2001
Name and mailing address of the ISA/ Swedish Patent Office Box 5055, S-102 42 STOCKHOLM Facsimile No. +46 8 666 02 86	Authorized officer Sture Elnäs /itw Telephone No. +46 8 782 25 00

INTERNATIONAL SEARCH REPORT

International application No.
PCT/SE 01/00512

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	WO 0048367 A (NOKIA WIRELESS ROUTERS,INC.), 17 August 2000 (17.08.00), abstract --	1-15
A	HAJEK,B.et al."SCHEDULING NONUNIFORM TRAFFIC IN A PACKET SWITCHING SYSTEM WITH LARGE PROPAGATION DELAY." In:Information Theory,IEEE Transactions on On:pp. 358-365,March 1995 vol.41.part 4. -- --------	1-15

INTERNATIONAL SEARCH REPORT
Information on patent family members

01/10/01

International application No.
PCT/SE 01/00512

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
US 6055578 A	25/04/00	EP 0919090 A US 5894559 A WO 9807257 A	02/06/99 13/04/99 19/02/98
US 5854900 A	29/12/98	WO 9745984 A	04/12/97
US 5850525 A	15/12/98	WO 9737464 A	09/10/97
US 4412326 A	25/10/83	NONE	
WO 0048367 A	17/08/00	AU 2342100 A	29/08/00